

APPLICATION OF NETWORK VOICE TO NAVY AND DOD TELECOMMUNICATIONS

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ABSTRACT

The application of network voice technology to Navy and DoD telecommunications offers possibilities for highly integrated communications services including voice, data, and video while maintaining interoperability with legacy communication systems. The need for network voice technology can be understood by considering the level of voice integration being achieved in contemporary shipboard backbone networks. Voice transport across shipboard Asynchronous Transfer Mode (ATM) networks being installed under the Navy's Information Technology for the 21st Century (IT-21) initiative is limited to using the ATM Forum's Circuit Emulation standard. Using Internet Protocol (IP) and Application Programming Interfaces (API) for conducting H.323 multimedia conferencing sessions to handle shipboard voice traffic is an attractive option for overcoming this limitation.

In this paper, we identify options available for developing network voice applications for military communication systems. A specific example of applying network voice technology for developing a solution for interoperating with legacy shipboard communications is discussed.

INTRODUCTION

Voice transport across shipboard ATM networks being installed under the Navy's Information Technology for the 21st Century (IT-21) [1] initiative is limited to using the ATM Forum's Circuit Emulation standard. Circuit Emulation [2] of Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) trunks does not allow dynamic routing of voice circuits. Instead, Circuit Emulation uses Permanent Virtual Circuits with a constant bit rate to pass voice traffic between predetermined endpoints.

A Standard exists for voice over native ATM and more standards are under study. However, the ATM switches and edge devices selected for shipboard IT-21 installations do not support this standard and none of these standards address Voice over IP [3][4]. Every workstation needs an ATM or ISDN interface card to use applications based on this standard.

To be cost effective, any voice transport solution needs to run on workstations without requiring additional hardware and cabling. This means frame-based networking to the desktop. For example, of the approximately 600 drops being installed on USS ENTERPRISE for IT-21 only about 100 drops bring ATM to the desktop.

In addition to the workstation issue, any solution needs to have mechanisms for easily supporting legacy communication systems.

There are no viable plans to replace the Half-Duplex Tactical radio voice communications between ships and aircraft. There will be a continuing need to support this type of voice service into the foreseeable future.

Support for tactical radio voice service is required to make backfit installations usable. It costs too much to replace these systems wholesale.

A solution based on emerging open standards for supporting voice communications and conferencing over computer networks can be deployed on IT-21 computer workstations and it is possible that the solution can be tailored to support tactical radio requirements [5].

TARGET ARCHITECTURE

Figure 1 depicts a target architecture for a Network Voice Terminal. Voice transport over the ATM backbone and Local Area Network (LAN) segments

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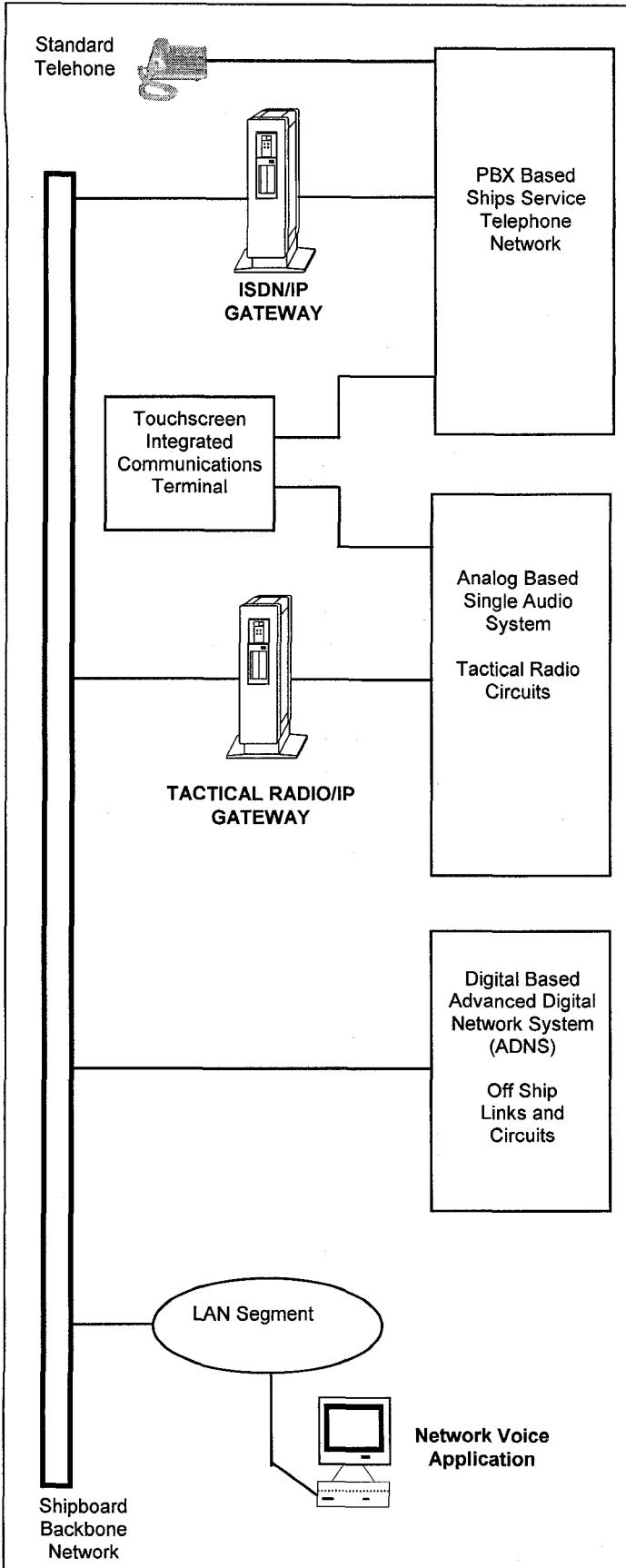


Figure 1. Target Architecture for the Network Voice Terminal.

occurs as H.323 multimedia conference sessions [6]. Voice transport with ISDN switches and legacy tactical radio equipment depends on Gateways to provide interfaces between the networks.

The Network Voice Terminal is simply a software application which brings up a user interface providing warfighters with the same functionality as a touchscreen Integrated Communications Terminal (ICT), but without the expense of additional hardware and cabling. Figure 2 shows a notional subsystem software architecture diagram for a Network Voice Terminal.

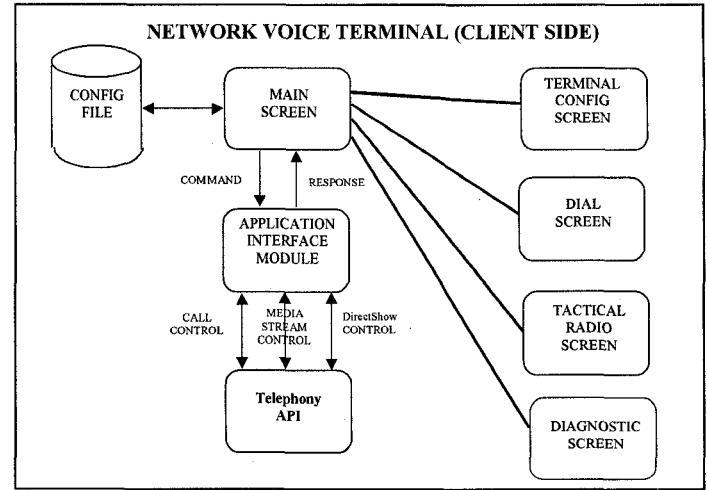


Figure 2. Notional Subsystem Software Architecture Diagram.

A number of options are available upon which to base the development and implementation of a Network Voice Terminal concept. Microsoft Corporation provides development kits and Application Programming Interfaces (APIs) based on its Telephony (TAPI) and Netmeeting technologies. These programming interfaces provide access to standards-based voice call and conferencing services. Other companies such as DataBeam Corporation, Brooktrout Technology, and Natural Microsystems also provide software development kits for creating standards-based voice call and conferencing applications. Another network voice implementation, the Interactive VOice eXchange (IVOX) utilizes military standard low data rate voice compression techniques to support point-to-point and group (multicast) voice communications on computer networks [7][8][9]. IVOX has already been successfully utilized in a number of military communications systems and demonstrations including strategic command and control and tactical wireless and satellite communication networks.

Some of these toolkits provide support for data and video conferencing in addition to the voice services

which offers possibilities for highly integrated communications services beyond what can be achieved by voice-only systems. Related standards encompassed by these toolkits include the H.323 network voice and conferencing standard, T.120 data conferencing standard, and Internet Location Service (ILS) and Lightweight Directory Access Protocol (LDAP) for using standards-based directory services to perform such functions as dynamic user-to-IP address resolution.

Other related technology includes Internet Engineering Task Force (IETF) working groups and standards such as the Resource Reservation Protocol (RSVP) for dynamically signaling application Quality-of-Service (QoS) requirements in computer networks, Multimedia Session Control (MMUSIC), IP Multicast and a new working group focused on Internet telephony. DoD participation in these working groups coupled with exploratory development and implementation can lead to fully standards based solutions which can potentially meet military requirements for functionality, security, and compatibility with legacy systems. Security is an particularly important area where government participation in the standards working groups can make the standards more effective for military communication systems.

In the next section we discuss applying network voice technology for developing a solution for interoperating with shipboard legacy tactical radio communication circuits.

LEGACY TACTICAL RADIO SIGNALING

Legacy tactical radio signaling for US Navy ships is part of the Single Audio System [10]. The Single Audio System was designed to provide remote radio circuit access to users distributed throughout a ship. In its most modern form, it consists of an SA-2112 Secure Voice or Red Switch and user station equipment including TA-970/U telephone sets and C-10276 Channelizers.

Name	Direction	Description
PTT	User to Radio	Push-To-Transmit, a continuous signal that keys transmitter when asserted.
Hksw	User to Switch	Hookswitch, a continuous signal requesting service asserted when handset is off cradle
MS	User to Crypto Device	Mode Select, a momentary signal requesting a change to cypher mode when asserted one way and requesting a change to plain mode when asserted the other way. Idle otherwise.
MI	Crypto Device to all Users	Mode Indicate, a continuous signal indicating the crypto mode (plain or cypher) for the transmit side of the circuit.
Det	Crypto Device to all Users	Cypher Detect, a continuous signal indicating that a valid transmission is being decrypted by the crypto device.

Table 1. Tactical Radio Signal Descriptions

The signaling to support tactical radio circuits consists of call progress signaling and crypto control signaling. Descriptions for tactical radio signals are summarized in Table 1. Mode Indicate and Cypher Detect signals must be multicast to all users connected to the tactical radio circuit or net. Transmit and receive audio also needs to be multicast or conferenced to all users connected to the net.

TECHNICAL APPROACH

A network-based voice system offers a great deal of flexibility and simplicity to accommodate these signaling needs and to provide effective gateways to legacy tactical voice systems, particularly when compared to typical analog switched, or even digital switched, telephony systems.

The H.323 and related standards offer a number of possibilities for incorporating these signals as part of the network voice system. It may be possible to extend the functionality of the signaling portion of the emerging H.323 standard or to encapsulate the needed signals in specialized data messages. This approach isolates the user specific signaling to the application layer of the communication system making it independent of the underlying network (and cabling) infrastructure.

The tactical radio signaling will be captured and analyzed using Specification Description Language. This is the same technique used in defining ISDN's Link Access Protocol Using the D-Channel (LAPD) of Q.921. The signals used for tactical radio evolved without the benefit of this type of analysis and problems such as failing to properly clear channels when disconnecting from radio calls have resulted from it.

The present plan is to create a prototype Network Voice Terminal in 1999. This effort will be used to validate the technical approach for handling the tactical radio signaling described above, develop a security and key management solution, and continue to explore the technology area and outline an approach for applying network-based voice communications in military systems.

REFERENCES

- [1] Vice Admiral Arthur Cebrowski and John J. Garstka, "Network-Centric Warfare Its Origin and Future," U.S. Naval Institute Proceedings, January 1998.
- [2] ATM Forum, "Circuit Emulation Service Interoperability Specification Version 2.0," af-vtoa-0078.000, January 1997.

[3] ATM Forum, "Voice and Telephony Over ATM to the Desktop Specification," af-vtoa-0083.000, May, 1997.

[4] ATM Forum, "Voice and Telephony Over ATM - ATM Trunking using AAL1 for Narrowband Services Version 1.0," AF-VTOA-0089.000, July 1997.

[5] Microsoft Corporation, "IP Telephony with TAPI 3.0," White Paper, 1997. Available at www.microsoft.com

[6] International Telecommunications Union, "H.323 Line Transmission of Non-Telephone Signals," Geneva, 1996.

[7] Brian Adamson, "IVOX, the Interactive Voice eXchange Application," NEWLINK Global Engineering Technical Report 101, September 1995. Available at www.ngec.com.

[8] Brian Adamson and Joe Macker, "IVOX - The Interactive Voice eXchange Application," MILCOM 96 Conference Proceedings, 1996.

[9] Brian Adamson and Joe Macker, "IVOX - The Interactive Voice eXchange Application," Naval Research Laboratory Report 5520-96-9805, February 1996.

[10] Naval Electronic Systems Engineering Activity, "Single Audio System Student Handbook," St Inigoes, MD, 1989.